Telecommunications Theory

The explosive growth of telecommunications traffic continues to generate ever-increasing demands for radio spectrum while greatly increasing the loading of telecommunications networks, both wireless and wireline. Yet the radio spectrum is a limited resource. In response to these realities, new radio technologies are being developed and implemented to use spectrum more efficiently and effectively. Also, the basic paradigm of radio spectrum management is beginning to move away from traditional, top-down frequency-assignment methods and is migrating toward autonomous, interference-limited technologies that allow dynamic reassignment of radio frequencies by systems that utilize intelligent algorithms for recognizing and avoiding interference. But to fulfill the promise of more autonomous, locally self-controlled spectrum use schemes, the effects of noise and interference on radio receiver performance must be thoroughly understood, and such knowledge must be focused on improvements in the performance of both existing and new networks. Tools to monitor the quality of audio and video information on communication channels also must be developed and used so that audio and video quality levels can be accurately adjusted in real-time to achieve maximal quality with minimal use of available bandwidth.

To achieve these goals for the U.S. government as well as the private sector, the Institute's Telecommunications Theory Division performs research in both wireless and wireline telecommunications, seeking to understand and improve telecommunications at the most fundamental levels of physics and engineering. Strong ongoing investigations are being maintained in the major areas of broadband wireless systems performance; short-range propagation modeling and measurements; noise and interference as critical limiting factors for advanced communication systems; audio and video quality assessment; advanced spectrum sharing concepts; and radio propagation.

Through technical publications, cooperative research and development agreements (CRADAs), and interagency agreements, ITS transfers the results of its work in all these technology areas to both the public and private sector, where the knowledge is transformed into better telecommunications for the United States, new and better products for consumers and the Government, and new opportunities for economic development and growth for the economy.

Areas of Emphasis

Audio Quality Research
The Institute conducts research and development leading to standardization and industry implementation of perception-based, technology-independent, quality measures for voice and other audio communication systems. Projects are funded by NTIA.

Effects of Radio Channel on Receiver Performance
The Institute, a recognized leader in radio channel measurement and modeling, is researching the effects of interference and noise on the performance of radio receivers and networks. Current work is focused on the effects of noise and interference as limiting factors in system performance. The project is funded by NTIA.

Video Quality Research
The Institute develops perception-based, technology-independent, video quality measures and promotes their adoption in national/international standards. Projects are funded by NTIA.

Audio Quality Research

Outputs

- Technical publications and presentations on new research results.
- Measurements and estimates of speech and audio quality and algorithm performance.
- Algorithms and data supporting speech and audio coding and quality assessment.

Digital coding and transmission of speech and audio signals are enabling technologies for many telecommunications and broadcasting services including voice over Internet protocol (VoIP) services, cellular telephone services, and digital audio broadcasting systems. Speech and audio signals can be coded and transmitted at low bit-rates with good fidelity. In addition, coded speech and audio signals can be packetized for transmission, thus sharing network bandwidth or radio spectrum with other data streams.

It is important to note that digital coding and transmission of speech and audio involves compromises and trade-offs among at least five basic factors: signal quality, transmission bit-rate, robustness to transmission errors and losses, coding and transmission delay, and coding and transmission algorithm complexity. For a simple example of one trade-off, consider receive-side buffering of data packets delivered by a best-effort network. Increasing the size of a buffer can reduce the number of packets lost (desirable), but this will also increase the algorithm delay (undesirable). For a complete system of coding and transmission, all five factors will generally come into play. For any given application, joint optimization with respect to these five factors can be an attractive but elusive goal.

The ITS Audio Quality Research Program works to identify, develop, and characterize new techniques that will increase relative quality or robustness or will reduce relative bit-rate, delay, or complexity of speech and audio coding and transmission algorithms. In addition, the Program seeks to develop techniques that aid in attaining a desired optimum balance between the five factors. Integral to this effort is the development of more effective and efficient ways to characterize speech and audio quality, since this can be the most intangible of the five factors. The ultimate result of these efforts is better sounding, more reliable, more efficient telecommunications and broadcasting services.

The robustness of digital coding and transmission algorithms is critical in applications that use lossy channels such as those associated with wireless systems and those provided by the Internet. In FY 2006, Program staff evaluated the robustness of a specific family of speech coding algorithms. Example results are shown in Figure 1. This figure shows how one measure of speech quality drops as the percentage of lost data packets increases. The results shown in blue correspond to a more robust algorithm and the results shown in red correspond to a less robust algorithm (quality drops more quickly as packet loss increases).

In FY 2006, Program staff performed an extensive evaluation of three objective estimators of speech quality. These algorithms process digital speech signals and generate estimates of perceived speech quality. These estimates can then be compared with actual perceived speech quality values that have been obtained through formal subjective listening tests using the same speech signals. If an objective estimator provides speech quality estimates that largely agree with the actual perceived speech quality values, then that estimator can be readily used. But if the agreement is poor, then the estimator can be used only with great caution.

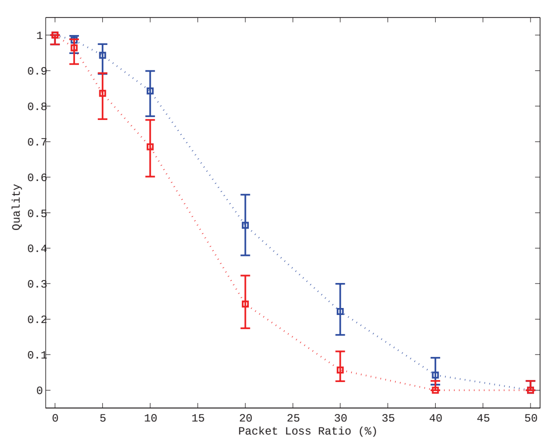

Figure 1. Examples showing how speech quality drops as packet loss ratio increases for two speech coding algorithms.

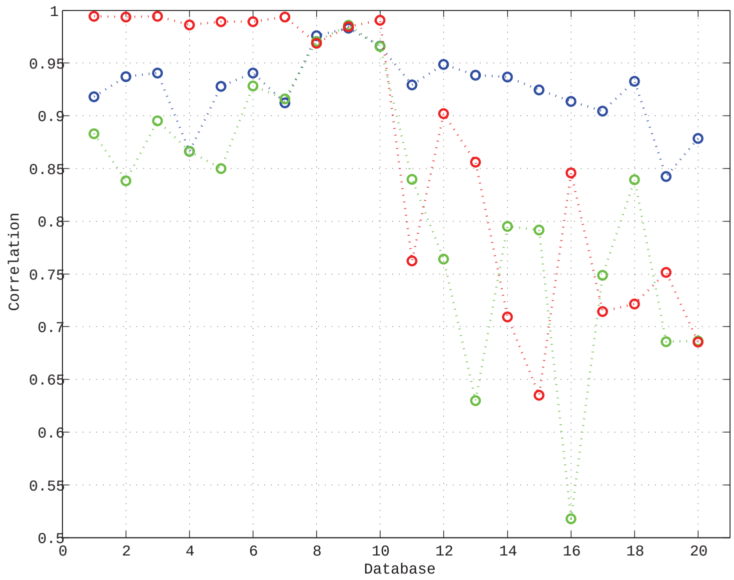

Figure 2. Example performance of three different objective estimators of speech quality on 20 different speech databases.

In our evaluation we measured agreement through the use of a correlation value. A value of 1.0 indicates perfect agreement, and smaller values indicated lesser agreement. Our evaluation used 20 different databases of digitally recorded speech signals. Together these databases include a wide variety of speech signals and systems under test. Example results are given in Figure 2. This figure shows correlation values for the three objective estimators of speech quality. The results in blue describe an estimator that has consistent, moderately good agreement (correlation above 0.9 in most cases). The results in green describe an estimator that has a widely varying agreement. The results in red describe an estimator that has very good agreement on the first ten databases (correlations mostly near 1.0), but a much lower and less consistent agreement on the second ten databases. These results suggest that this third estimator has potential if it can be successfully generalized for operation on databases 11-20 with agreement similar to that found on databases 1-10.

Quantization is a pervasive component in speech and audio coding. It enables concise digital representations, but it always induces some quantization error or quantization noise. In FY 2006, Program staff developed and characterized a new generalized technique that can reduce the noise associated with certain quantization processes when particular classes of signals are used. This technique is centered on the use of a pseudoinverse that requires a small amount of prior knowledge of the signal at design time. Given this information, the pseudoinverse may be constructed from an appropriate adaptive filter. In operation, the filter portion of the pseudoinverse adapts so that the statistics of its input-output difference signal are matched to the statistics of the associated quantizer noise. In certain cases, this results in partial cancellation of the original quantization noise.

Throughout FY 2006, Program staff continued with significant speech quality testing using both objective and subjective techniques. These tests support both this program and other ITS programs. Staff continued to transfer program results to industry, Government, and academia by means of technical publications, lectures, laboratory demonstrations, and poster presentations. Staff also completed a significant number of peer reviews for technical journals and conference proceedings. Program staff supported telecommunications standards development through research efforts, technical exchanges, and a formal written technical contribution. Program publications, technical information, and other program results are available at http://www.its.bldrdoc.gov/audio.

Recent Publications

S.D. Voran, “Listening-time relationships in a subjective speech quality experiment,” in *Proc. 5th International MESAQIN (Measurement of Speech and Audio Quality in Networks) Conference*, Prague, Czech Republic, Jun. 2006.

S.D. Voran, “Reducing quantization error by matching pseudoerror statistics,” in *Proc. of the 12th IEEE Digital Signal Processing Workshop*, Grand Teton National Park, Wyoming, Sep. 2006.

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Effects of Radio Channel on Receiver Performance

Outputs

- Analytic and discrete receiver models.
- Correlations between radio channel characteristics and receiver performance.
- Uncertainty analysis of communications system measurements.

Telecommunications play a vital role in providing services deemed essential for modern life. Many systems providing these services use radio links composed of a transmitter, a receiver, and a channel separating the two. The radio channel is often the primary impediment to fast and reliable operation of these systems. Understanding how the channel affects those systems is crucial to the advancement and regulation of telecommunications. Hence, ITS has historically directed considerable research towards radio channel characterization. This project expands this focus to include quantifying the effects of the radio channel on receiver performance.

Receiver performance is degraded by a number of radio channel phenomena, such as attenuation, multipath, man-made noise, and signals from other radio links. Various combinations of these factors affect each system uniquely. For example, personal communications services (PCS) systems operating outdoors in the 1900-MHz band are primarily compromised by time-varying multipath introduced by buildings and terrain. Wireless local area network (WLAN) systems operating indoors in the 900-, 2400-, and 5800-MHz industrial, scientific, and medical (ISM) bands contend with man-made noise from microwave ovens, signals from cordless phones, and multipath introduced by reflections from walls, ceilings, floors, and objects within the room.

Practical limitations prevent us from testing the effects of the radio channel phenomena on all legacy or proposed radio systems. Hence, the goal of this effort is to identify, model, and characterize a small number of radio systems and use these to predict the effects of the radio channel on others. The results of this work are of vital importance in the development of telecommunications policy. For example, the results can be used to predict how interference introduced by new spectrum engineering methods such as spectral overlay will impact legacy systems.

The effects of the radio channel on receiver performance are quantified by correlating radio link performance metrics to radio channel characteristics, as shown in Figure 1 below. As an example, bit error rate (a performance metric) can be correlated to interfering signal power (a channel characteristic) as is shown in NTIA Reports TR-05-419,[1] TR-05-429,[2] and TR-06-437[3] on the interference potential of ultrawideband signals. These correlations are verified in three ways, i.e., mathematical analysis, software simulation, and laboratory measurements, when possible, to ensure reliability. The laboratory

1 M. Cotton, R. Achatz, J. Wepman, and B. Bedford, "Interference potential of ultrawideband signals — Part 1: Procedures to characterize ultrawideband emissions and measure interference susceptibility of C-band satellite digital television receivers," NTIA Report TR-05-419, Feb. 2005.

2 M. Cotton, R. Achatz, J. Wepman, and P. Runkle, "Interference potential of ultrawideband signals — Part 2: Measurement of gated-noise interference to C-band satellite digital television receivers," NTIA Report TR-05-429, Aug. 2005.

3 M. Cotton, R. Achatz, J. Wepman, and R. Dalke, "Interference potential of ultrawideband signals — Part 3: Measurement of ultrawideband interference to C-band satellite digital television receivers," NTIA Report TR-06-437, Feb. 2006.

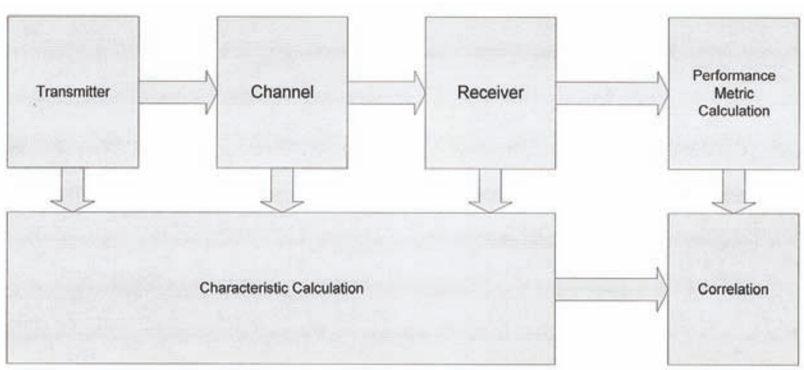

Figure 1. Conceptual diagram of approach which correlates radio link characteristics to radio receiver performance.

measurements can be executed with commercial off-the-shelf equipment or vector signal generators and analyzers downloaded by transmitter and receiver emulation codes. Considerable effort is vested in evaluating the uncertainties associated with working from the finite sample sets of measurements and simulations.

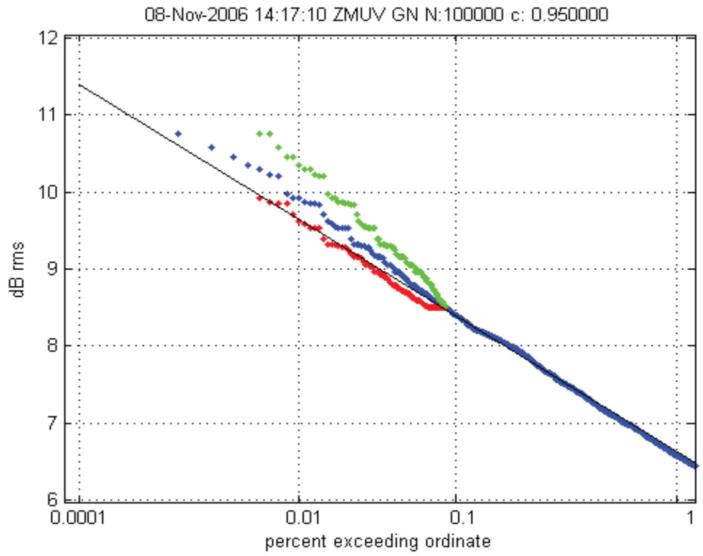

Figure 2. Confidence interval applied to low probabilities of an APD built from 100,000 samples of simulated zero-mean, unit-variance Gaussian noise. The solid line is the theoretical APD. The three dotted lines represent the measured APD flanked by the upper and lower 95% confidence interval values.

Success depends on project engineers' familiarity with communications signal processing methods, mathematical theory of probability, and statistical analysis. Engineers must also understand radio-frequency measurement methods along with radio channel measurement and analysis techniques.

Currently, efforts are focused on the mathematical analysis and software simulation of a linearly modulated radio system operating within a Gaussian noise channel. Linear modulation was chosen because it includes a wide range of commonly used modulations such as pulse amplitude modulation, phase shift keying, and quadrature amplitude modulations. Linear modulations are also used by low power radio links most vulnerable to interference, such as satellite links.

Considerable progress has been made in FY 2006. Major sections of the forthcoming report, "Effects of the radio channel on receiver performance. Part 1: Analytic model" by R.J. Achatz and R.A. Dalke, have been written. These sections include a mathematical description of the linearly modulated transmitter/receiver pair, including signal, noise, and receiver components, derivations of signal and noise amplitude probability distributions (APD), power spectral density (PSD), and average power characteristics, and extensive appendices which help the reader understand the communications analysis techniques involved. This report is expected to be completed in FY 2007. Work on "Part 2: Discrete model," which will include software simulation code, will begin upon publication of Part 1.

In addition, a comprehensive review of the statistical considerations for communications systems measurement or simulation is nearing completion. This work will culminate in the NTIA Report, "Statistical considerations for communication systems measurement" by R.A. Dalke, expected to be published in FY 2007. This work addresses the calculation of the uncertainties associated with computing performance metrics and statistical signal characteristics from the finite data sets provided by measurement. Uncertainties associated with measurement of APDs, PSDs, and average power have been completed. Uncertainties associated with bit error statistics are now being developed. An example of this work is shown in Figure 2.

In FY 2006, ITS formed an important alliance with NTIA's Office of Spectrum Management (OSM) regarding the application of this work. OSM is interested in including pertinent results in its "Best Practices Handbook," which will be used by all spectrum regulatory bodies in the United States Federal Government. In particular, they are interested in using these results as guidance for the assessment of the effect the radio channel has on modern signal processing methods such as error correction coding.

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Video Quality Research

Outputs

- Digital video quality measurement technology.
- Journal papers and national/international video quality measurement standards.
- Technical input to development of U.S. policies on advanced video technologies.
- A national objective and subjective digital video quality measurement laboratory.

Objective metrics for quantifying the performance of digital video systems (e.g., direct broadcast satellite, digital television, high definition television, video teleconferencing, telemedicine, internet, and cell phone video) are required by end-users and service providers for specification of system performance, comparison of competing service offerings, network maintenance, and use optimization of limited network resources. The goal of the ITS Video Quality Research project is to develop the required technology for assessing the performance of these new digital video systems and to actively transfer this technology to other Government Agencies, end-users, standards bodies, and the telecommunications industry, thereby producing increases in quality of service that benefit all end-users and service providers.

To be accurate, digital video quality measurements must be based on perceived "picture quality" and must be made in-service. This is because the performance of digital video systems is variable and depends upon the dynamic characteristics of both the input video and the digital transmission system. To solve this problem, ITS has continued to develop new measurement paradigms based upon extraction and comparison of low bandwidth perception-based features that can be easily communicated across the telecommunications network.

These new measurement paradigms (now commonly known throughout the world as "reduced reference" measurements) have received four U.S. patents (with one additional patent pending), been adopted as the North American Standard for measuring digital video quality (ANSI T1.801.03-2003), been included in two International Recommendations (ITU-T Recommendation J.144, and ITU-R Recommendation BT.1683), and are currently being used by hundreds of individuals and organizations worldwide.

To facilitate the transfer of ITS-developed video quality metrics (VQMs) into the private sector, ITS has developed three software tools that run under both the Windows and Linux operating systems. Using these new software tools, users and service providers can quantify the digital video quality of their networks using methods standardized by

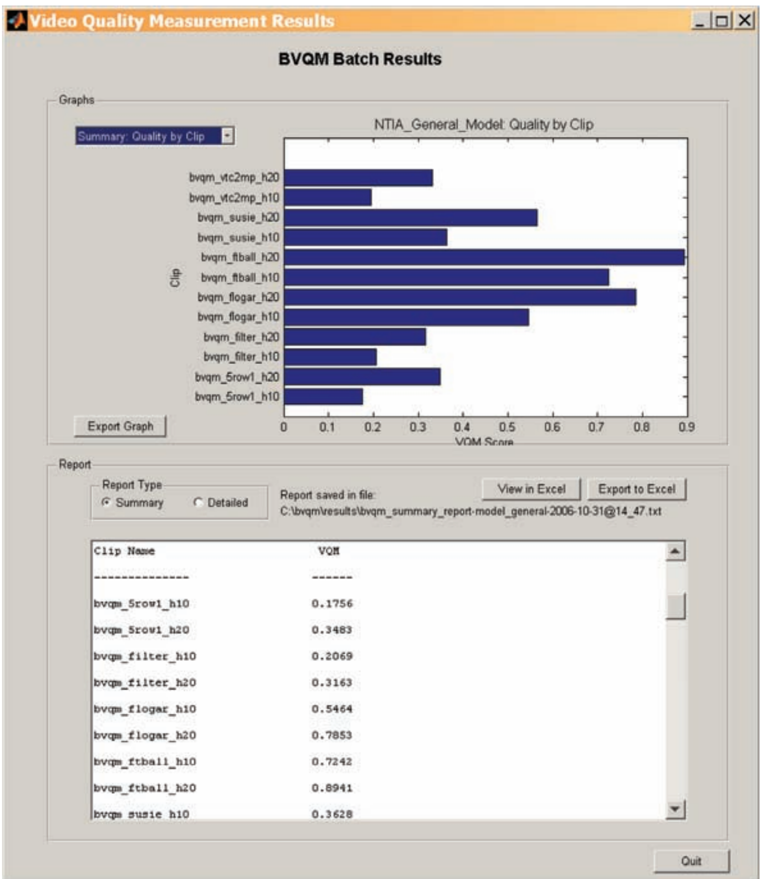

Figure 1. Example snapshot of the BVQM results screen showing summary results for the processed video clips.

the American National Standards Institute (ANSI) and the International Telecommunication Union (ITU).

The first tool, called the “Laboratory VQM Tool,” is useful for bench testing of video systems. For this tool, video from the source and destination ends of a video system must be present at a single PC for analysis.

The second tool, called the “In-Service VQM (IVQM) Tool,” requires two PCs, one located at the source end and the other located at the destination end of a digital video transmission system. The two PCs communicate their reduced reference features via the Internet, producing in-service end-to-end video quality monitoring results.

The third tool, developed in FY 2006, is called the “Batch VQM (BVQM) Tool.” The BVQM tool allows the user to perform Graphical User Interface (GUI) based batch mode processing of many captured video streams, or files.

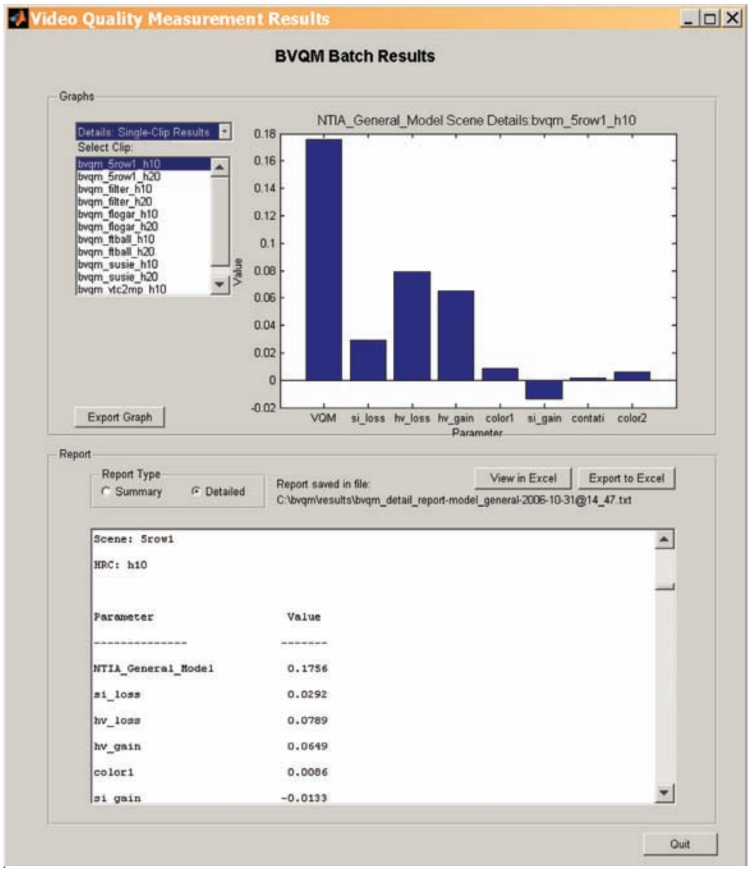

Figure 2. Example snapshot of the BVQM results screen showing detailed results for a user-selected video clip.

Figures 1 and 2 give example screen snapshots of the BVQM results screen after processing multiple video files. The user can display summary results for the processed video clips (Figure 1) or detailed results for a user-selected video clip (Figure 2). Other options (not shown) include displaying results by video system (i.e., averaged over all video scenes) or by video scene (i.e., averaged over all video systems). Summary and detailed reports of the video calibration (e.g., gain and level offset, spatial scaling/registration, temporal registration) and VQM scores/parameters can be saved in either text or Microsoft Excel® formats.

During FY 2006, 126 new Cooperative Research and Development Agreements (CRADAs) were implemented with U.S. companies/individuals and 75 new Evaluation License Agreements (EVAs) were implemented with foreign companies/individuals. These CRADAs and EVAs provide companies with an easy mechanism for evaluating ITS video quality measurement technology and software before signing commercial licensing agreements. As a result of this arrangement, five companies signed commercial licensing agreements with ITS in FY 2006.

Recent Publications

M. McFarland, M.H. Pinson, and S. Wolf, “Batch video quality metric (BVQM) user’s manual,” NTIA Handbook HB-06-441, Sep. 2006.

M.H. Pinson and S. Wolf, “Reduced reference video calibration algorithms,” NTIA Report TR-06-433a, Jul. 2006.

M.H. Pinson and S. Wolf, “In-service video quality metric (IVQM) user’s manual,” NTIA Handbook HB-06-434a, Jul. 2005.

Further information can be found on the Video Quality Research home page at http://www.its.bldrdoc.gov/n3/video.

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